



Avaya Solution & Interoperability Test Lab

Configuring Session Initiated Protocol over Port Network Address Translation for Avaya 4602 SIP IP Telephones using the Kagoor VoiceFlow 200 Application Layer Gateway - Issue 1.0

Abstract

These Application Notes describe how to configure the Kagoor VoiceFlow 200 Application Layer Gateway (ALG) to handle Avaya 4602 Session Initiated Protocol (SIP) Telephone traffic through a Cisco 2621 router configured for Port Network Address Translation (PNAT). The sample VoIP reference configuration depicted in this document consists of an Avaya Converged Communications Server located at the service provider data center and Avaya 4602 SIP IP Telephones located at the enterprise site.

1. Introduction

These Application Notes describe how to configure the Kagoor VoiceFlow 200 Customer Premise Equipment (CPE) based Application Layer Gateway (ALG) to handle Avaya 4602 Session Initiated Protocol (SIP) Telephone traffic through a Cisco 2621 WAN router configured for PNAT. The sample VoIP reference configuration depicted in this document consists of an Avaya Converged Communications Server located at the service provider data center and Avaya 4602 SIP IP Telephones located at the enterprise site. All Avaya 4602 SIP Telephones are configured to register with the VoiceFlow 200 instead of the Avaya Converged Communications Server. The VoiceFlow 200 appears to be the Avaya Converged Communications Server to the SIP endpoints, and appears to be the SIP Endpoints to the Converged Communications Server.

The Kagoor VoiceFlow 200 is a VoIP-aware network device that processes, measures, and manipulates VoIP packets in the IP and application layers. VoIP calls are tracked using the common VoIP protocols (H.323, SIP, MGCP). VoiceFlow devices transparently perform wire-speed manipulations on signaling and media streams. The VoiceFlow 200 device can be located at any location at the enterprise and does not require special modifications or configurations to existing physical network infrastructure. The VoiceFlow 200 can operate in any of the following three modes:

- Maintenance mode – used for maintenance operations such as configuring the device IP parameters. When operating in maintenance mode, packets pass through the VoiceFlow system without being intercepted.
- Normal mode – when operating in this mode, the VoiceFlow is fully functional, classifying and processing packets based on the defined configuration.
- Safe mode – unlike maintenance or normal, this mode is enabled automatically upon critical system malfunction. Safe mode is used as a recovery mechanism with a limited amount of management capabilities.

Figure 1 illustrates the components of the Avaya SIP configuration. The Kagoor VoiceFlow 200 ALG located at the enterprise location is connected to the Cisco Catalyst switch and is configured as the SIP endpoints Registrar and Proxy server, which is managing all of the VoIP traffic, while not affecting the data traffic. The VoiceFlow 200 ALG acts as a VoIP proxy between the endpoints on the private side and the Avaya Converged Communications Server on the public side. Since the VoiceFlow 200 needs to be configured as the Registrar and SIP Proxy for the SIP endpoints, the DHCP and HTTP server configurations for the 4602 SIP IP Telephones must reflect the private IP address and port number of the Kagoor VoiceFlow 200 ALG as the Registrar and SIP Proxy IP addresses. Please refer to [1] for more information on the DHCP and HTTP server configurations for Avaya 4602 SIP IP Telephones.

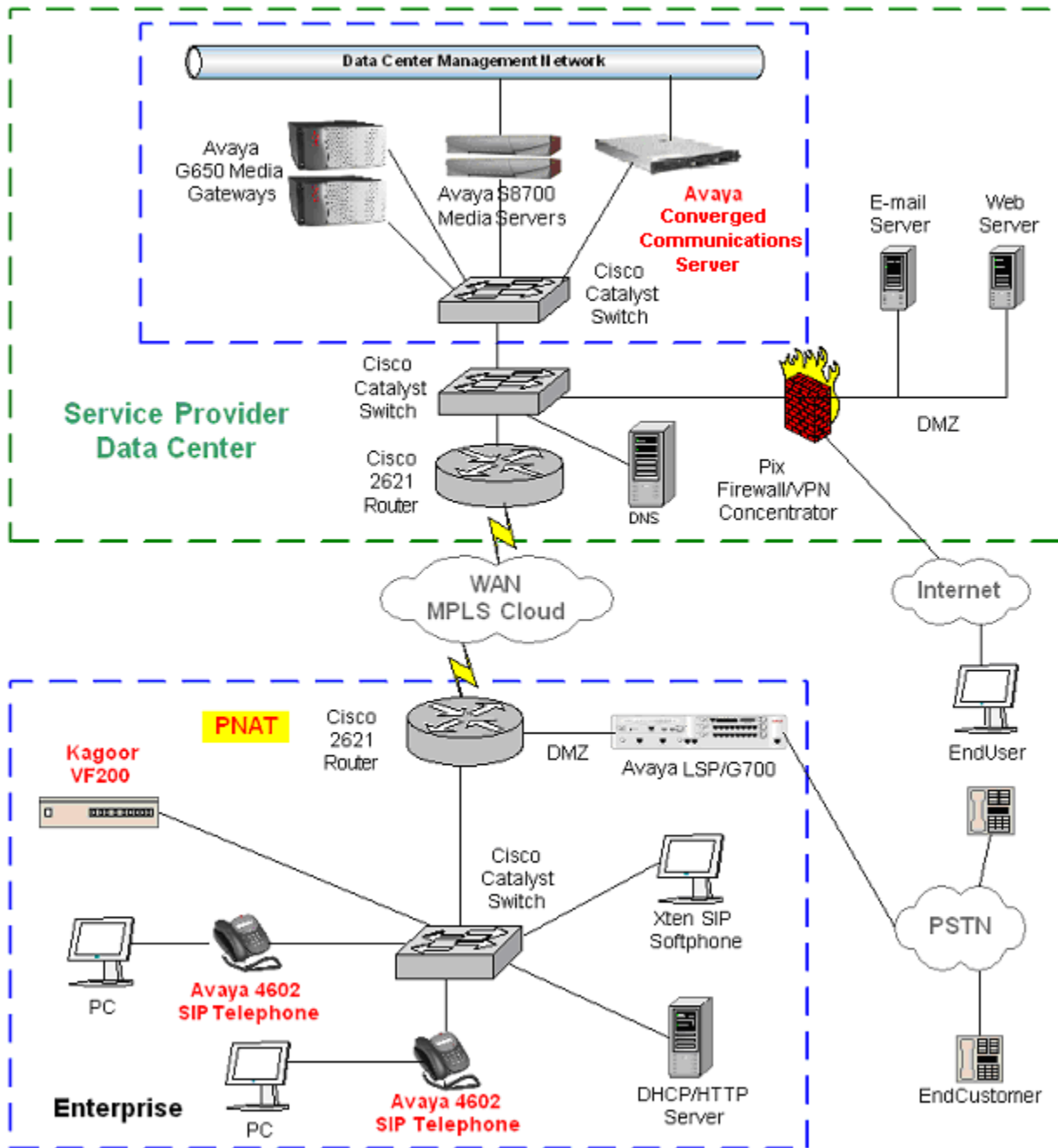


Figure 1: Avaya SIP PNAT Configuration

Note: These Application Notes address the Kagoor VoiceFlow 200 and Cisco 2621 WAN router configurations. Please consult the appropriate user guides for more information on configuring the other network devices depicted in **Figures 1 and 2**.

Figure 2 illustrates the IP address configuration for the relevant Avaya SIP network components shown in **Figure 1**. Two Virtual LANs (VLANs) were created to keep the voice and data traffic separate from each other. The voice VLAN (VLAN 101) was assigned subnet 10.1.1.0/24 and the data VLAN (VLAN 102) was assigned subnet 10.1.2.0/24. The Cisco 2621 WAN router was used as a gateway for both networks by extending an IEEE 802.1Q trunk from the Cisco catalyst switch to the Cisco 2621 WAN router.

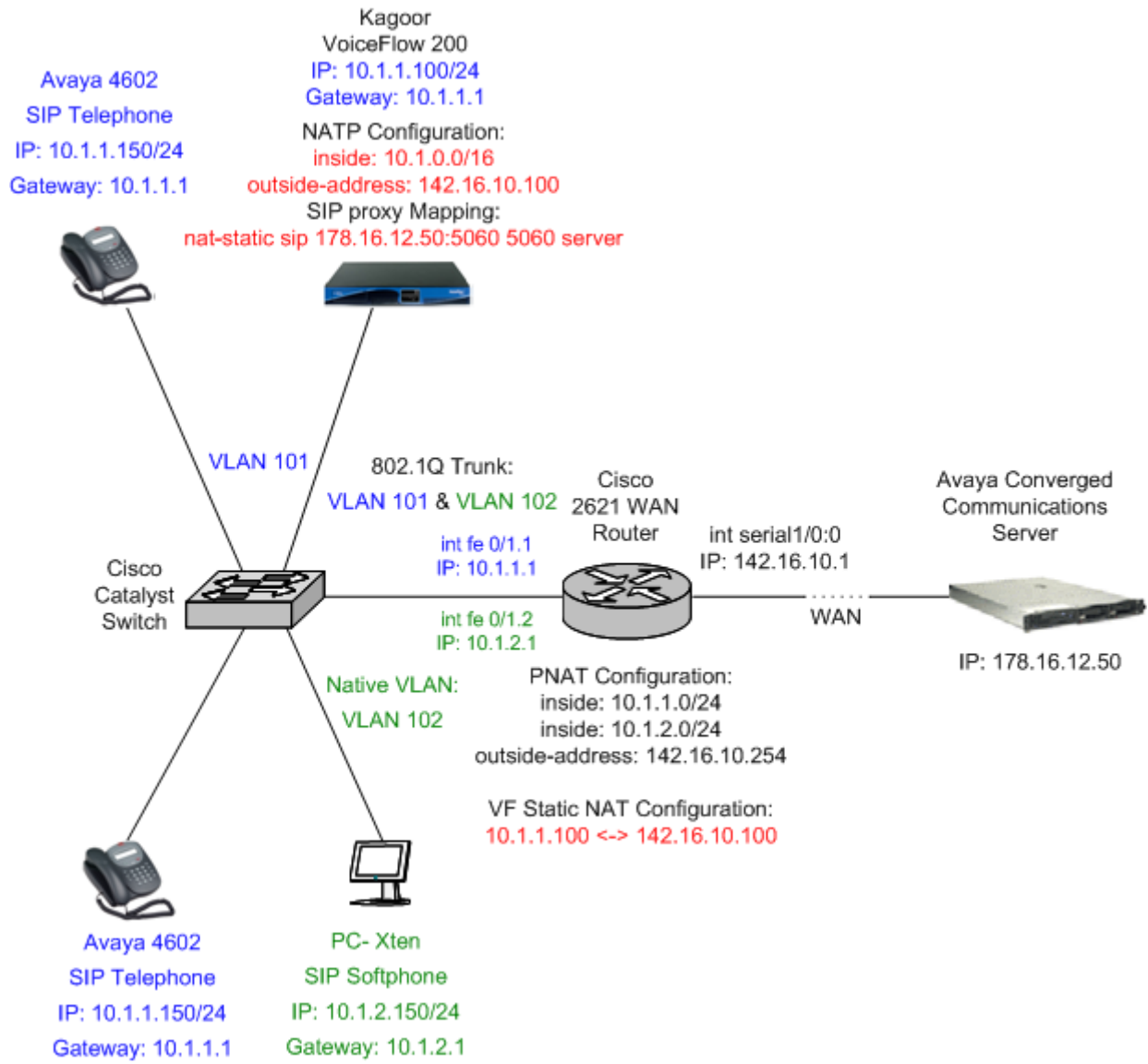


Figure 2: IP Address Configuration

Figure 3 illustrates the SIP registration message flow between an Avaya 4602 SIP Telephone and the Avaya Converged Communications Server through the Kagoor VoiceFlow 200 and the Cisco 2621 router configured for PNAT. The first time the Avaya 4602 Telephone sends out the “REGISTER” message, it does not include authentication in the SIP Session Description Protocol (SDP) header. Since Avaya Converged Communications Server requires authentication by default, it sends a “401 Unauthorized” message back to the Telephone, which causes it to send out another “REGISTER” message with authentication credentials in the SDP header.

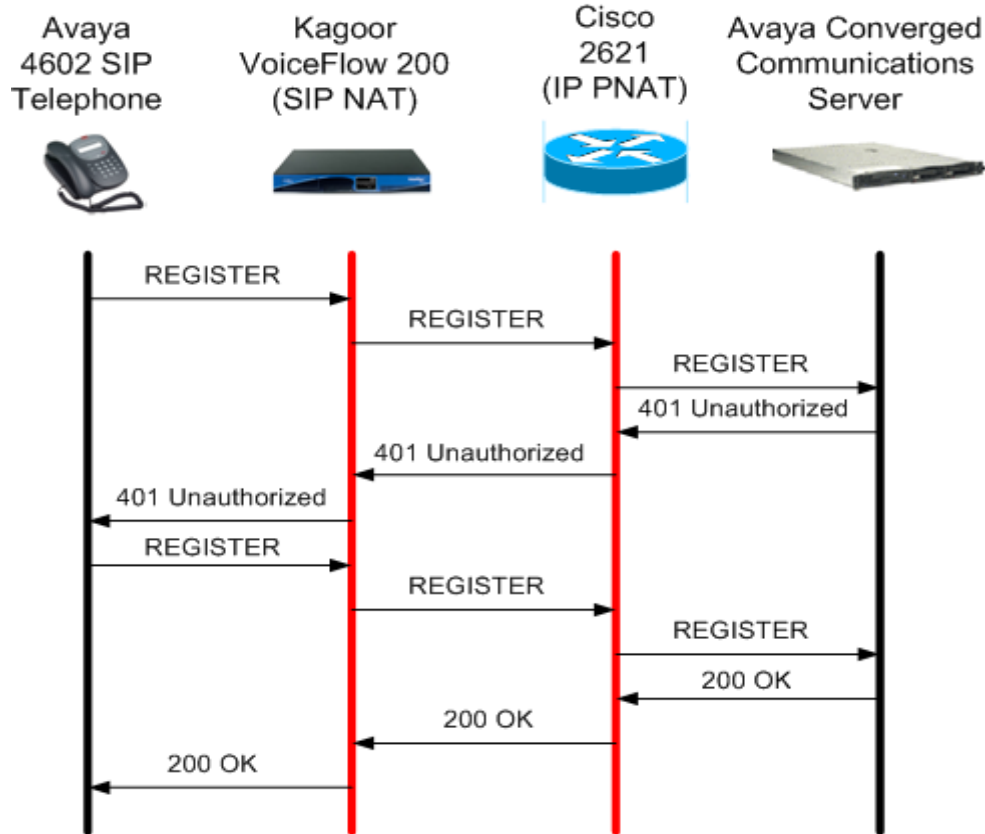


Figure 3: SIP Registration Messages

Figure 4 illustrates a sample call flow between two SIP endpoints through the Kagoor VoiceFlow 200 and the Cisco 2621 router configured for PNAT during normal call setup and tear down. Although the Avaya SIP endpoints were configured as Avaya Communication Manager Off-PBX Telephone Integration and Mobility (OPTIM) endpoints in these Application Notes, the call flow assumes that Avaya Communication Manager is not involved.

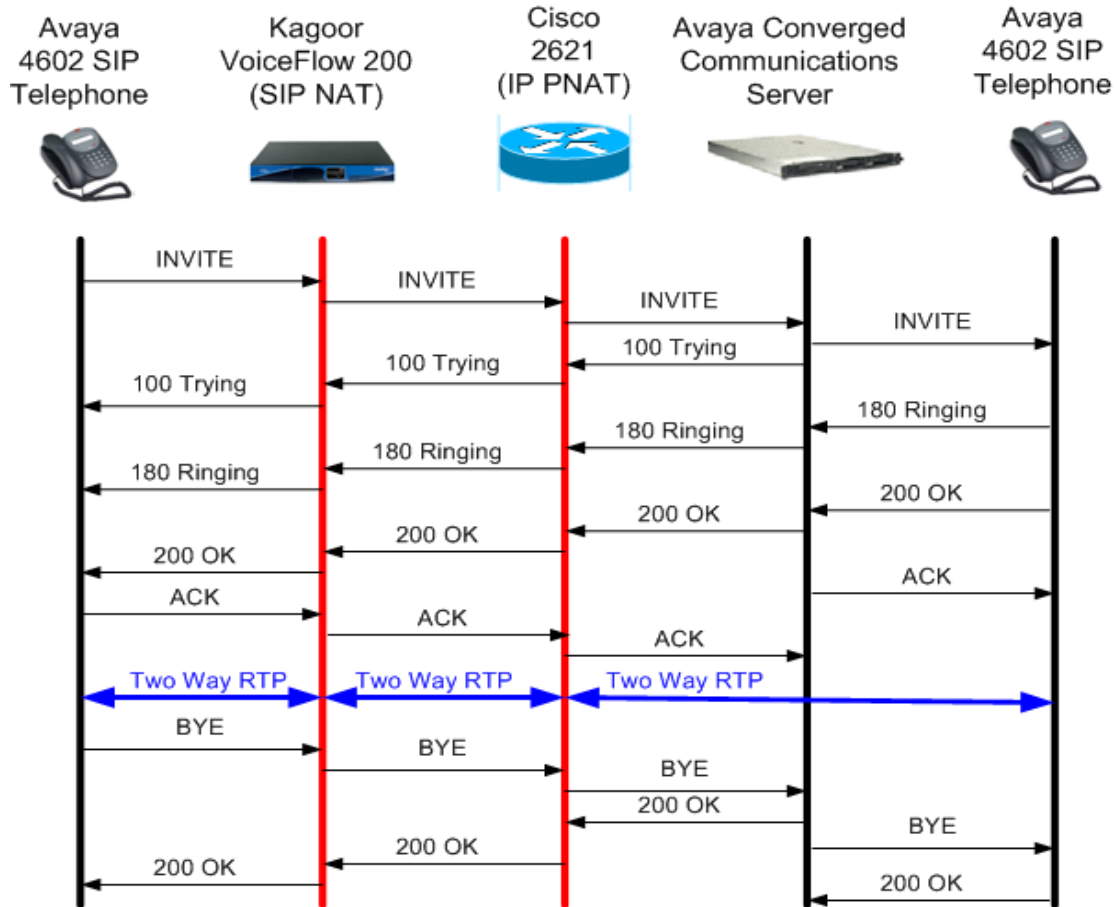


Figure 4: SIP Call Flow Messages Without OPTIM

Note: The call message flow in **Figure 4** assumes that Avaya Communication Manager OPTIM endpoints are not configured. Note that OPTIM is currently a requirement for Avaya Converged Communications Server and the example above is only shown to illustrate the messages going through the Kagoor VoiceFlow 200 during SIP NAT traversal.

2. Hardware and Software Validated

Hardware and Software	Version
Avaya Converged Communications Server	2.1.1
Avaya 4602 SIP Telephones	1.1
Kagoor VoiceFlow 200 ALG	5.3.1.011avmc
Xten SIP Softphone	X-Lite 2.0 (build 1099)
Cisco 2621 Routers	12.2(8)T4
Cisco 3600 Routers	12.2(8)T5
DHCP/HTTP Servers: Microsoft Windows 2000 Server	5.00.2195 (SP2)

Table 1: Hardware and Software Versions

3. Kagoor VoiceFlow 200 Configuration

The following configuration was done through the Kagoor VoiceFlow 200 command line interface (CLI) via an RS-232 serial cable from the console port.

Step	Description
1.	<p>Enter “privileged” mode to input configuration and management commands. Enter the “privilege” mode password.</p> <pre>Welcome to VoiceFlow ALG-V/OS [ip missing]> enable Password: ALG-V/OS [ip missing]#</pre>
2.	<p>Enter “maintenance” mode to configure the system IP address:</p> <pre>ALG-V/OS [ip missing]# mode maintenance</pre>
3.	<p>Assign an IP address and default gateway to the VoiceFlow system:</p> <pre>ALG-V/OS [ip missing]# ip 10.1.1.100/24 gateway 10.1.1.1 Routing table entries were cleared Note: IP address changed! ALG-V/OS 10.1.1.100#</pre>
4.	<p>Enter VoIP NAT configuration:</p> <pre>ALG-V/OS 10.1.1.100# blade nat ALG-V/OS 10.1.1.100(nat)#</pre>
5.	<p>Enable Keep-Alive Control and configure the endpoint keep-alive interval. The default interval is 30 seconds. This value can be modified by entering “endpoint-keepalive-period x”, where x = 10 to 600 seconds.</p> <pre>ALG-V/OS 10.1.1.100(nat)# keepalive-control ALG-V/OS 10.1.1.100(nat)# endpoint-keepalive-period 120</pre>
6.	<p>Set the private network configuration. The prefix/mask (10.1.0.0/16) was used to allow IP endpoints on the voice (IP Telephones) and data (Xten SIP Softphones) networks to be processed by the VoiceFlow. Source IP addresses outside of the (10.1.0.0/16) prefix/mask are ignored:</p> <pre>ALG-V/OS 10.1.1.100(nat)# inside 10.1.0.0/16</pre>

Step	Description
7.	<p>Assign a global IP address to the VoiceFlow 200 corresponding to the outside IP address defined for the VoiceFlow 200 system by the data NAT outside the LAN (Cisco 2621 WAN router):</p> <pre data-bbox="391 373 1271 401">ALG-V/OS 10.1.1.100(nat)# outside-address 142.16.10.100</pre>
8.	<p>Assign a UDP port range to be used for port allocation of Real Time Protocol (RTP) traffic. Since all SIP Telephones are configured as Avaya Communication Manager OPTIM endpoints (configuration not shown), make sure the UDP port range matches the Avaya Communication Manager IP Network Region configuration assigned to the Avaya Converged Communications Server trunk group:</p> <pre data-bbox="391 653 1320 680">ALG-V/OS 10.1.1.100(nat)# udp-port-range dynamic 2048-3028</pre>
9.	<p>Configure the SIP NAT traversal mapping to the IP address of the Avaya Converged Communications Server:</p> <pre data-bbox="391 827 1401 940">ALG-V/OS 10.1.1.100(nat)# nat-static sip 178.16.12.50:5060 5060 server ALG-V/OS 10.1.1.100(nat)# exit ALG-V/OS 10.1.1.100#</pre> <p>Note: The Avaya SIP Telephones must be configured with the private IP address of the VoiceFlow 200 as the Registrar and Proxy IP addresses. The VoiceFlow 200 appears to be the Avaya Converged Communications Server to the SIP endpoints, and appears to be the SIP Endpoints to the Avaya Converged Communications Server.</p>
10.	<p>Enter “normal” mode to process VoIP traffic:</p> <pre data-bbox="391 1234 906 1262">ALG-V/OS 10.1.1.100# mode normal</pre>
11.	<p>Save the Kagoor VoiceFlow 200 configuration:</p> <pre data-bbox="391 1373 1271 1400">ALG-V/OS 10.1.1.100# copy running-config startup-config</pre>

4. Cisco 2621 WAN Router Configuration

The following configuration was done through the Cisco 2621 WAN router command line interface (CLI) via an RS-232 serial cable from the console port.

Step	Description
1.	<p>Enter router privileged mode by entering the privileged mode password:</p> <pre>C2621WAN> enable Password: Cisco2621#</pre>
2.	<p>Enter router configuration mode:</p> <pre>C2621WAN# configure terminal Enter configuration commands, one per line. End with CNTL/Z. C2621WAN(config)#</pre>
3.	<p>Configure the Cisco 2621 WAN physical line properties. The T1 configuration is shown below:</p> <p>The following commands show how to configure the T1 interface parameters:</p> <pre>C2621WAN(config)# controller T1 1/0 C2621WAN(config-controller)#</pre> <p>Set up the frame format and line encoding format for the T1 link. The T1 controller below is configured for Extended Super Frame (ESF) frame format and Binary 8-Zero Substitution (B8ZS) line coding.</p> <pre>C2621WAN(config-controller)# framing esf C2621WAN(config-controller)# linecode b8zs</pre> <p>Next, a valid clock source must be configured. The available clock choices are “line”, “internal”, and “loop-timed”. The Cisco 2621 WAN router is set to get clock from the line.</p> <pre>C2621WAN(config-controller)# clock source line</pre> <p>A list of timeslots (DS0s) for voice channels must be configured for the T1 controller. This can be done using the command “channel-group <channel-no> timeslots <timeslot-list> speed <48 56 64>”.</p> <pre>C2621WAN(config-controller)# channel-group 1 timeslots 1-24 speed 64 C2621WAN(config-controller)# no shutdown C2621WAN(config-controller)# exit C2621WAN(config)#</pre>

Step	Description
4.	Configure the Ethernet interface connected to the G700: <pre> C2621WAN(config)# interface Ethernet0/0 C2621WAN(config-if)# ip address 150.100.3.1 255.255.255.0 C2621WAN(config-if)# ip nat outside C2621WAN(config-if)# no shutdown C2621WAN(config-if)# exit </pre>
5.	Configure the Ethernet Private sub-interfaces: <pre> C2621WAN(config)# interface Ethernet0/1 C2621WAN(config-if)# no shutdown C2621WAN(config-if)# exit C2621WAN(config)# interface Ethernet0/1.1 C2621WAN(config-subif)# encapsulation dot1Q 101 C2621WAN(config-subif)# ip address 10.1.1.1 255.255.255.0 C2621WAN(config-subif)# ip helper-address 10.1.2.61 C2621WAN(config-subif)# ip nat inside C2621WAN(config-subif)# exit C2621WAN(config)# C2621WAN(config)# interface Ethernet0/1.2 C2621WAN(config-subif)# encapsulation dot1Q 102 native C2621WAN(config-subif)# ip address 10.1.2.1 255.255.255.0 C2621WAN(config-subif)# ip nat inside C2621WAN(config-subif)# exit C2621WAN(config)# </pre>
6.	Configure the serial interface to the ISP MPLS network: <pre> C2621WAN(config)# interface serial 1/0:0 C2621WAN(config-if)# ip address 142.16.10.1 255.255.255.0 C2621WAN(config-if)# ip nat outside C2621WAN(config-if)# encapsulation ppp C2621WAN(config-if)# no shutdown C2621WAN(config-if)# exit C2621WAN(config)# </pre>
7.	Enable OSPF routing on Avaya G700 Media Gateway and public interfaces: <pre> C2621WAN(config)# router ospf 1 C2621WAN(config-router)# network 142.16.10.0 0.0.0.255 area 0.0.0.0 C2621WAN(config-router)# network 150.100.3.0 0.0.0.255 area 0.0.0.1 C2621WAN(config-router)# exit C2621WAN(config)# </pre>
8.	Enable RIP on the private interface: <pre> C2621WAN(config)# router rip C2621WAN(config-router)# network 10.1.0.0 C2621WAN(config-router)# exit C2621WAN(config)# </pre>

Step	Description
9.	Configure port NAT (PNAT) for non-SIP devices: <pre data-bbox="391 302 1398 390">C2621WAN(config)# ip nat pool ovrlld 142.16.10.254 142.16.10.254 prefix-length 24 C2621WAN(config)#</pre>
10.	Configure access lists for network addresses to be considered for PNAT: <pre data-bbox="391 504 1430 653">C2621WAN(config)# ip nat inside source list 1 pool ovrlld overload C2621WAN(config)# access-list 1 deny host 10.1.1.100 C2621WAN(config)# access-list 1 permit 10.1.1.0 0.0.0.255 C2621WAN(config)# access-list 1 permit 10.1.2.0 0.0.0.255 C2621WAN(config)#</pre> <p data-bbox="298 695 1479 762">Note: The “access-list 1 deny host 10.1.1.100” command is needed to prevent the IP address of the Kagoor VoiceFlow 200 device from being considered for PNAT on the edge router.</p>
11.	Configure a static NAT entry for the Kagoor VoiceFlow 200 global IP outside address: <pre data-bbox="391 875 1284 963">C2621WAN(config)# ip nat inside source static 10.1.1.100 142.16.10.100 C2621WAN(config)#</pre>
12.	Copy the running configuration to the startup configuration. <pre data-bbox="391 1077 789 1255">C2621WAN(config)# exit C2621WAN# write memory Building configuration... [OK] C2621WAN#</pre>

5. Verification Steps

SIP calls over PNAT can be verified by placing calls between Avaya 4602 SIP Telephones located at the enterprise site or between a PSTN phone and SIP Telephones at the enterprise site. The NAT parameters in the VoiceFlow 200 ALG can be displayed with the “**show nat parameters**” command. The “**show active endpoints**” command shows the active NAT bindings in the Kagoor VoiceFlow 200 ALG. The command to check the NAT bindings in the Cisco 2621 WAN router is “**show ip nat translations**”.

6. Conclusion

The Kagoor VoiceFlow 200 CPE based ALG can be successfully configured to handle SIP traffic over PNAT originating from Avaya 4602 SIP IP Telephones registered to an Avaya Converged Communications Server.

7. References

[1] “*Configuring Microsoft DHCP and HTTP Servers for Avaya 4602 SIP IP Telephones*”, Issue 1.0

[2] “*VoiceFlow Series V/OS release 5.3 CPE Border Controller Installation and User Manual*”, Part. No. DO-0024-01

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